PATENT APPLICATION

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of:

Renaud CUNY et al.

Application No.: New Application

Filed: Concurrently Herewith

Attorney Dkt. No.: 60091.00288

For: METHOD AND SYSTEM FOR CONTROLLING A REAL-TIME COMMUNICATIONS SERVICE

CLAIM FOR PRIORITY UNDER 35 USC § 119

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

April 20, 2004

Sir:

The benefit of the filing dates of the following prior foreign application filed in the following foreign country is hereby requested for the above-identified patent application and the priority provided in 35 U.S.C. §119 is hereby claimed:

Finnish Patent Application No. 20031912 filed on December 29, 2003 in Finland

In support of this claim, a certified copy of said original foreign application is filed herewith.

It is requested that the file of this application be marked to indicate that the requirements of 35 U.S.C. §119 have been fulfilled and that the Patent and Trademark Office kindly acknowledge receipt of this document.

Please charge any fee deficiency or credit any overpayment with respect to this paper to Counsel's Deposit Account No. 50-2222.

Respectfully submitted,

Douglas H. Goldhush
Registration No. 33,125

Customer No. 32294
SQUIRE, SANDERS & DEMPSEY LLP
14TH Floor
8000 Towers Crescent Drive
Tysons Corner, Virginia 22182-2700
Telephone: 703-720-7800
Fax: 703-720-7802

DHG:lls

Enclosure: Priority Document

PATENTTI- JA REKISTERIHALLITUS
NATIONAL BOARD OF PATENTS AND REGISTRATION

Helsinki 12.1.2004

E T U O I K E U S T O D I S T U S
P R I O R I T Y   D O C U M E N T

Hakija Applicant	**Nokia Corporation** **Helsinki**
Patenttihakemus nro Patent application no	**20031912**
Tekemispäivä Filing date	**29.12.2003**
Kansainvälinen luokka International class	**H04L**

Keksinnön nimitys
Title of invention

"Method and system for controlling a real-time communications service"
(Menetelmä ja järjestelmä reaaliaikaisen tiedonsiirtopalvelun kontrolloimiseksi)

Täten todistetaan, että oheiset asiakirjat ovat tarkkoja jäljennöksiä Patentti- ja rekisterihallitukselle alkuaan annetuista selityksestä, patenttivaatimuksista, tiivistelmästä ja piirustuksista.

This is to certify that the annexed documents are true copies of the description, claims, abstract and drawings originally filed with the Finnish Patent Office.

Pirjo Kalla
Tutkimussihteeri

Maksu 50
Fee 50 EUR

Maksu perustuu kauppa- ja teollisuusministeriön antamaan asetukseen 1027/2001 Patentti- ja rekisterihallituksen maksullisista suoritteista muutoksineen.

The fee is based on the Decree with amendments of the Ministry of Trade and Industry No. 1027/2001 concerning the chargeable services of the National Board of Patents and Registration of Finland.

Osoite: Arkadiankatu 6 A Puhelin: 09 6939 500 Telefax: 09 6939 5328
P.O.Box 1160 Telephone: + 358 9 6939 500 Telefax: + 358 9 6939 5328
FIN-00101 Helsinki, FINLAND

L2

METHOD AND SYSTEM FOR CONTROLLING A REAL-TIME COMMUNICATIONS SERVICE

Field of the Invention

The present invention relates to real-time communications services in communication systems.

Background of the Invention

Particularly in the third generation (3G) mobile communications systems, a public land mobile network (PLMN) infrastructure may be logically divided into a core network (CN) 9,10,11,12 and an access network (AN) infrastructures 5,6,7,8, as illustrated in Fig 1. The access network AN may be called base station subsystem (BSS) 8 for GSM and radio network subsystem (RNS) or radio access network (RAN) 5,6,7 for UMTS. In the technical specifications of third generation partnership project (3GPP), the core network CN is logically divided into a circuit switched (CS) domain 9, a packet switched (PS) domain 10,11 and an IP multimedia subsystem (IMS) 12. The CS domain refers to the set of all the CN entities offering "CS type of connection" for user traffic as well as all the entities supporting the related signalling. A "CS type of connection" is a connection for which dedicated network resources are allocated at the connection establishment and released at the connection release. A "PS type of connection" transports the user information using packets so that each packet can be routed independently from the previous one. Example of the PS domain may be the GPRS (General Packet Radio Service), and the typical entities may include serving GPRS support node (SGSN) and gateway GPRS support node (GGSN). The IP multimedia subsystem comprises all CN elements for provision of multimedia services. The IP multimedia subsystem IMS utilizes the PS domain to transport multimedia signalling and bearer traffic.

Push-to-talk over Cellular (PoC) is an overlay speech service in a mobile cellular network where a connection between two or more parties is established (typically) for a long period but the actual radio channels in the air interface are activated only when somebody is talking. This corresponds the usage of the traditional radiotelephones where the used radio frequency is agreed between the parties (e.g. military/police radios, LA radios) of permanently set (walkie-talkie type of radios) and whenever somebody wants to talk she/he presses the tangent, which activates the radio transmission in the se-

lected channel. The traditional radiotelephone services are simplex by their nature so that only one party (the one who is pressing the tangent) can talk at a time. More specifically, in a voice communication with "push-to-talk, release-to-listen" feature, a call is based on the use of a pressel (PTT, push-to-talk switch) in a telephone as a switch: by pressing a PTT the user indicates his desire to speak, and the user equipment sends a service request to the network. Alternatively, a voice activity detector (VAD) or any suitable means can be used instead of the manual switch. The network either rejects the request or allocates the requested resources on the basis of predetermined criteria, such as the availability of resources, priority of the requesting user, etc. At the same time, a connection is established also to a receiving user, or users in the case of group communication. After the voice connection has been established, the requesting user can talk and the other users can listen to. When the user releases the PTT or in the case of traffic inactivity, the event is detected in the network, and the resources may be released and/or talk item may be granted to another user. Thus, the resources are reserved only for the actual speech transaction or speech item, instead of reserving the resources for a "call".

Modern cellular networks, especially in the GSM/GPRS/UMTS network evolution, include new packet-mode (e.g. IP) voice and data services. Push-to-talk over Cellular (PoC) service can be provided as a packet-based user or application level service so that the underlying communications system only provides the basic connections (i.e. IP connections) between the group communications applications in the user terminals and the group communication service. The PoC communication service can be provided by a communication server system while the client applications reside in the user equipments or terminals. Examples of this approach are disclosed in co-pending U.S. patent applications 09/835,867; 09/903,871; and 10/160,272; and in WO 02/085051.

With the PoC service, first the connection(s) between the parties is established typically via the packet switched (PS) mobile network, e.g. a packet switched (PS) core network. In practice, this means that a Voice over IP (VoIP) (group or one-to-one) call is set up between the parties. However, as described above, the difference to conventional phone call is that the radio channel of the subscribers is activated only when somebody needs to talk and released when nobody is talking.

The PoC service is a practical solution for the cases when the parties need to talk relatively rarely but whenever somebody needs to talk, the connection has to be activated fast and easily (e.g. when giving instructions to the members of a hunting team in the forest or to a crane driver in a construction site). Because in this type of applications the calls are typically long but the voice activity is low, it is essential to release the bearer (e.g. radio channels) while nobody is talking in order to save the radio and network capacity and terminal batteries. On the other hand, the bearer resources should be available with as small delay as possible when voice activity again starts.

Disclosure of the Invention

An object of the invention is to decrease the delay associated with voice transmission in a real-time media communication.

The object is achieved by the invention defined in the attached independent claims. Preferred embodiments of the invention are defined in the sub claims.

An aspect of the invention is a method of controlling a real-time media session, comprising

sending first signalling from a first user equipment via a serving access network of the first user equipment to a first media communication server in response to user's action during an established real-time media session,

sending second signalling from the first media communication server towards the first user equipment,

sending third signalling from the first media communication server towards second user equipment,

sending immediately following the first signalling and/or the second signalling and/or the third signalling, dummy media traffic from the first media communication server towards the first user equipment, in order to trigger a dedicated channel setup for the first user equipment and/or the second user equipment in the serving access network of the first user equipment prior to an actual user media stream from the first user equipment begins.

An aspect of the invention is a method of controlling a real-time media session, comprising

establishing a real-time media session between first user equipment and second user equipment via a serving access network of the first user equipment, via at least a first media communication server, and via a serving

access network of the second user equipment,

sending, by the media communication server or a support node in a packet switched core network during inactive periods of the real-time media session, dummy media towards at least one of the first and second user equipment, in order to reset an inactivity timer of a common channel state in the serving access network of the respective user equipment and to thereby prevent the respective user equipment from going to an idle state.

An aspect of the invention is a media communication server for providing real-time media sessions between user equipments located in one or more access networks, wherein

the media communication server is configured to receive first signalling sent by first user equipment via a serving access network of the first user equipment in response to user's action during an real-time media session established between the first user equipment and a second user equipment,

the media communication server is configured to send second signalling towards the first user equipment upon receiving said first signalling,

the media communication server is configured to send third signalling towards the second user equipment upon receiving said first signalling,

the media communication server is configured to send, immediately following the first, second and/or third signalling, dummy media traffic towards the first and/or second user equipment, in order to trigger a dedicated channel setup for the first and/or second user equipment in the respective serving access network prior to an actual user media stream from the first user equipment begins.

In an embodiment of the invention a media communication server is configured to send dummy media traffic to first and/or second user equipment only if the session inactivity prior to first signalling exceeded a certain threshold, in order to limit the amount of unnecessary dummy data sent.

An aspect of the invention is a support node for a packet switched core network, wherein

the support node is configured to establish a real-time media connection between a user equipment located in a radio access network, and a media communication server,

the support node is configured to send, during inactive periods of the real-time media connection, dummy media towards the user equipment, in order to reset an inactivity timer of a common channel state in the radio access

network and to thereby prevent the respective user equipment from going to an idle state.

An aspect of the invention is user equipment for a communication system, wherein

the user equipment is configured to establish a real-time media session via an access network and a media communication server,

the user equipment is configured to send a first signalling via the access network to the media communication server in response to user's action during the established real-time media session, and

the user equipment is configured to send, immediately following the first signalling, dummy media traffic to the media communication server, in order to trigger a dedicated channel setup for the user equipment in the access network of the first user equipment prior to an actual user media stream begins.

In an embodiment of the invention the user equipment is configured to send dummy media traffic to the media communication server only if the session inactivity time prior sending the first signalling exceeds a certain threshold, in order to limit the amount of unnecessary dummy data sent.

The invention is based on sending dummy data (e.g. a dummy message) in order to maintain a dedicated channel during inactive periods of a real-time media session or to trigger an early dedicated-channel setup in an access network. The invention prevents user equipments that are logged on to a real-time media (e.g. PoC) session to go to radio resource idle state and therefore it avoids potential long extra delays during real time media (e.g. PoC) service usage. The invention further allows sending and receiving user equipments to set up their dedicated channels (DCH) already during the start to talk procedure of the transmitting user equipment, which in turn potentially reduces end-to-end delays during the conversation.

Brief Description of the Drawings

The above and other objects, features and advantages of the present invention will become more apparent in light of the following detailed description in conjunction with the drawings, in which

Figure 1 illustrates a communication system having a radio access network RAN, CS and PS core networks, and a PoC server,

Figure 2 is a block diagram illustrating functional blocks of a PoC server according to an example embodiment of the invention,

Figure 3 is a block diagram illustrating basic blocks of user equipment according to an example embodiment of the invention,

Figure 4 illustrates the various states of user equipment UE in WCDMA,

Figure 5 is a signaling diagram illustrating an example of a signaling flow for maintaining an active state of user equipments,

Figure 6 is a flow diagram illustrating an example of operation of a PoC server or a support node according to an embodiment of the invention,

Figure 7 is a signaling diagram illustrating an example of a signalling flow for setting up a media communication,

Figure 8 is a flow diagram illustrating an example of the operation of the UE in accordance with the principles of the present invention,

Figure 9 is a flow diagram illustrating an example of the operation of a PoC server in accordance with principles of the present invention, and

Figure 10 is a signalling diagram illustrating an example of a signalling flow for a communication event where a previous speaker stops speaking and a previous recipient starts speaking.

Detailed description

The present invention is applicable to communications systems enabling real-time media sessions between end users. The real-time data may include real-time audio (e.g. speech), real-time video, or any other real-time data, or combination thereof, i.e. real-time multimedia.

The present invention is especially applicable to communications system allowing packet-mode real-time data communication, such as IP packet communication between end users. Thus, the real-time data communication may be carried out between end user terminals over the Internet, for example.

The present invention offers a significant improvement for packet-mode speech communications. Voice over Internet Protocol (VoIP) enables a speech communication over an IP connection. In some embodiments of the invention, the IP voice communication method employed is the Voice over IP (VoIP), but the invention is not limited to this particular method.

As an example of a system environment to which the principles of the present invention may be applied to will be described with reference to

Figure 1. In Fig. 1, a Push-to-talk Over Cellular (PoC) server system is provided on top of the Packet Switched (PS) core network 10,11,12 in order to provide a packet mode (e.g. IP) voice, data and/or multimedia communication services to the User Equipment (UE) 1,2,3,4. An UE accessing the PS CN, and the PS core network itself, utilizes the services provided by the Radio network subsystem (RNS) or Radio access network (RAN) 5,6,7,8 to provide packet-mode communication between the UE and the PS CN subsystem. The multiple access method employed in the air interface in the RAN may be Time Division Multiple Access (TDMA), Frequency Division Multiple Access (FDMA), Code Division Multiple Access (CDMA), or a combination thereof. In the 3rd and higher generation mobile communications system the access method is primarily based on the CDMA. Further, because the traffic channels may have wide bandwidth, corresponding to user data rates e.g. up to 2 Mbits/s, such access may also to be referred as a Wideband CDMA (WCDMA).

Regarding to the PoC type services, examples of this concept are disclosed in co-pending U.S. patent applications 09/835,867; 09/903,871; 10/160,272; and in WO 02/085051. Conceptually, a packet based media communication system is provided on top of the mobile network in order to provide media communication services to the user equipment UE through the communication system. The media communication system may be embodied as a server system, and it is generally referred to as a media communication server herein. There may be a plurality of media communication servers 14,15. As illustrated in the example configuration of Figure 2, the media communication server may comprise control-plane functions CPF and user-plane functions UPF providing packet mode server applications that communicate with the communication client application(s) in the user equipment UE over the IP connections provided by the communication system. This communication includes signalling packets and voice or data communication packets. The CPF function is responsible for control-plane management of the group communication. This may include, for example, managing the user activity and creation and deletion of logical user-plane connections with an appropriate control protocol, such as Session Initiation Protocol (SIP). The user-plane function(s) UPF is responsible for distribution of the data or speech packets to the user terminals according to their group memberships and other settings. The UPF forwards traffic only between valid connections programmed by the CPF. In case of speech communication, it may be based on voice over IP (VoIP) protocol, and/or Real-time

Transport Protocol (RTP). It should be appreciated that the user plane operation relating to the data or speech traffic is not relevant to the present invention. However, the basic operation typically includes that all the data or speech packet traffic from a sending user is routed to the UPF which then delivers the packet traffic to the receiving user(s). The PoC server may include further entities, such as a register and a subscriber and group management function SGMF.

User equipment UE may be a wireless device, such as mobile user equipment, or it may be a device connected by a fixed connection, such as a dispatcher station. Herein a term user equipment and corresponding acronym UE is used for referring to any device or user equipment allowing the user to access network services.

As an exemplary embodiment, the user equipment UE, such as a Mobile Station MS, may have a PoC application on a user layer on top of the standard protocol stack used in the specific mobile communications system. An appropriate session control protocol, such as Session Initiation Protocol (SIP), may be used for the PoC control plane signaling. The voice communication may be based on IP communication (such as voice over IP, VoIP), and RTP (Real-time Transport Protocol, defined in RFC1889) may be employed to handle the voice packet (VoIP) delivery in the user plane. The SIP and RTP protocols employ the underlying Transmission Control Protocol (TCP), User Datagram Protocol (UDP) and IP protocols that further employ the physical layer resources, such as the radio resources. For example, the underlying connection in a mobile communication network may be based on a GPRS connection.

An example of a possible implementation of user equipment is illustrated in a simplified block diagram shown in Figure 3. An RF part 304 represents any radio frequency function and hardware required by a specific air interface employed. The actual implementation of the RF part 304 is not relevant to the present invention. A baseband signal processing 309 represents any baseband signal processing required in any specific implementation, such as an analog-digital (A/D) conversion of the analogue speech signal from the microphone 310, vo-encoding, IP packet building, frame building, deframing, IP packet debuilding, vo-decoding, a digital-analog (D/A) conversion of the received digital speech signal into an analog signal applied to a loudspeaker 311. A controller 305 controls operation of the RF unit 304 and the baseband

signal-processing unit 309. The controller 305 controls the signaling, both outband (SIP) and embedded, as well as IP packet building and debuilding. Start and stop of the speech items are set by the PTT switch 306 which can be replaced by any user-operated device, e.g. a voice activity detector (VAD). Such alternative mechanisms for starting and ending a speech item instead of the PTT are obvious to a person skilled in the art. A user interface may include a display 307 and a keyboard 308. It should be appreciated that the blocks illustrated in Figure 3 are functional blocks that can be implemented in a variety of different circuit configurations. For example, the baseband processing and the controller may be implemented in a single programmable unit (e.g. a CPU or a signal processor) or in a plurality of units. The operation according to the present invention is primarily related to the controller part of the MS, and the basic invention may be implemented as program modifications in the control program of the MS, for example. It should also be appreciated that the present invention is not intended to be restricted to mobile stations and mobile systems but the terminal can be any terminal having a speech communication capability. For example, the user terminal may be a terminal (such as a personal computer PC) having Internet access and a VoIP capability for voice communication over the Internet.

In the embodiment of Figure 3, the controller 305 comprises a media communication client application 301 (e.g. PoC client). The media communication client application 301 (e.g. PoC client) provides the respective communication service. For example, in the case of the PoC group communication, the client application 301 may maintain group information, such as group identification information and group membership information. The communication client 301 may also provide tools for group creation, for attaching (joining) to a group and for detaching from (leaving) the group, starting and ending the speech items, etc.

In PS core networks based on the GPRS or a like, the UE a) performs a GPRS attach procedure, and b) establishes a PDP context (i.e. a bearer) used for SIP signaling. This PDP context will remain active throughout the period the UE is connected to the PS CN, i.e. from the initial registration and at least until deregistration. As a result, the PDP context provides the UE with information that makes the UE able to construct an IP address. During establishment of a session, the UE establishes data stream(s) for media related to the session. Such data stream(s) may result in activation of additional

PDP context(s), i.e. bearers. Such additional PDP context(s) are established as secondary PDP contexts associated to the PDP context used for signalling. In other core network environments, other type of bearers may be used. It should be appreciated that the basic invention is basically independent from the type of the core network.

It should be appreciated that there are many applications of the Internet world that require the creation and management of a session, where a session is considered an exchange of data between an association of participants. The implementation of these applications is complicated by the practices of participants: users may move between endpoints, they may be addressable by multiple names, and they may communicate in several different media - sometimes simultaneously. Therefore, the present invention is not restricted to PoC services but can be applied for media flow management of such other applications as well.

Numerous protocols have been authored that carry various forms of real-time multimedia session data such as voice, video, or text messages. The Session Initiation Protocol (SIP, RFC 3261) general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols and without dependency on the type of session that is being established. SIP can be used with other IETF protocols to build up complete multimedia architecture. Typically, these architectures will include protocols such as the Real-time Transport Protocol (RTP) (RFC 1889) for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) (RFC 2326) for controlling delivery of streaming media, the Media Gateway Control Protocol (MEGACO) (RFC 3015) for controlling gateways to the Public Switched Telephone Network (PSTN), and the Session Description Protocol (SDP) (RFC 2327) for describing multimedia sessions.

It should be appreciated that VoIP or PoC are only examples of real-time media which the present invention can be applied to. It should also be appreciated that the type of the media session set up on the application level or the protocols used for controlling the media session on that level are not relevant to the basic invention. The present invention primarily relates to controlling the access bearers on the access network level, e.g. radio access bearers in the RAN.

In the following, example embodiments of the present invention will be described using 3GPP RAN (WCDMA) as an example of the access network.

In the 3GPP radio access environment, the user equipment may assume various protocol states. Figure 4 summarizes the mapping of UE states, including states in GSM, to the appropriate 3GPP and GSM specifications that specify the UE behavior. These specifications are incorporated herein by reference. However, only UE connected, CELL_DCH, CELL_FACH, and CELL_PCH are of interest in the following example embodiments of the invention.

After power on, the UE stays in Idle Mode until it transmits a request to establish an RRC (Radio Resource Control) Connection. In Idle Mode the connection of the UE is closed on all layers of the access stratum. In Idle Mode the UE is identified by non-access stratum identities such as an International mobile subscriber identity (IMSI), Temporary mobile subscriber identity (TMSI) and Packet TMSI (P-TMSI). In addition, the RNS has no own information about the individual Idle Mode UEs, and it can only address e.g. all UEs in a cell or all UEs monitoring a paging occasion.

The RRC Connected Mode is entered when the RRC Connection is established. The UE is assigned a radio network temporary identity (RNTI) to be used as UE identity on common transport channels. The transition to the RRC Connected Mode from the Idle Mode can only be initiated by the UE by transmitting a request for an RRC Connection. The event is triggered either by a paging request from the network or by a request from upper layers in the UE.

When the UE receives a message from the network that confirms the RRC connection establishment, the UE enters the CELL_FACH or CELL_DCH state of RRC Connected Mode. The RRC states within RRC Connected Mode reflect the level of UE connection and which transport channels that can be used by the UE.

In the CELL_DCH state, a dedicated physical channel is allocated to the UE in uplink and downlink, the UE is known on cell level according to its current active set, and dedicated transport channels, downlink and uplink shared transport channels, and a combination of these transport channels may be used by the UE.

The CELL_DCH-state is entered from the Idle Mode through the setup of an RRC connection, or by establishing a dedicated physical channel

from the CELL_FACH state. Transition to CELL_FACH state occurs when all dedicated channels have been released, which may be via explicit signaling (e.g. PHYSICAL CHANNEL RECONFIGURATION, Radio Bearer Reconfiguration, Radio Bearer Release, Radio Bearer Setup, Transport Channel Reconfiguration, etc.), or at the end of the time period for which the dedicated channel was allocated.

A transition from the CELL_DCH-state to the CELL_FACH state may occur after a predetermined period of inactivity. The period is monitored by means of an inactivity timer or timers. The period can be set to any value, typical value being 5 to 10 seconds.

In the CELL_FACH state, no dedicated physical channel is allocated to the UE and the UE continuously monitors a FACH in the downlink. The RAN may know the position of the UE on a cell level, i.e. according to the cell where the UE last made a cell update.

A transition from CELL_FACH to CELL_DCH state occurs, when a dedicated physical channel is established via explicit signaling (e.g. PHYSICAL CHANNEL RECONFIGURATION, RADIO BEARER RECONFIGURATION, RADIO BEARER RELEASE, RADIO BEARER SETUP, TRANSPORT CHANNEL RECONFIGURATION, etc.)

A transition from the CELL-FACH state may occur after a predetermined period of inactivity. The period is monitored by means of an inactivity timer or timers. The period can be set to any value, typical value being 5 to 10 seconds.

In the CELL_PCH state, no dedicated physical channel is allocated to the UE. The UE selects a PCH (Paging Channel) with a suitable algorithm, and uses discontinuous reception (DRX) for monitoring the selected PCH. Thus, the power consumption in the UE will be reduced. No uplink activity is possible. The position of the UE is known by UTRAN on cell level according to the cell where the UE last made a cell update in CELL_FACH state. A transition from the CELL_PCH state may occur into the Idle mode after a predetermined period of inactivity. The period is monitored by means of an inactivity timer or timers. The period can be set to any value, typical value being relatively long, e.g. 20-40 minutes.

Push-to-talk over Cellular (PoC) is a speech service in a mobile network where a connection between two or more parties is established (typically) for a long period but the actual radio channels in the air interface are

activated only when somebody is talking. With the PoC service the connections between the parties are typically established via a packet switched mobile network. In practice this means that a Voice over IP (VoIP) (group) call is set up between the parties. However, the difference to conventional phone call is that the radio channel of the subscribers is activated only when somebody needs to talk and released when nobody is talking. In more general terms, there is a streaming type real-time media signal having a session of long duration but requiring dedicated access resources (e.g. a DCH) only occasionally with fast set up times. There is a need for a method and means for controlling the activating and releasing the access releasing so that the fast set up time is achieved.

As noted above, a UE that does not transmit or receive any data (i.e. is inactive) will after some period of time go to radio resource control (RRC) idle state. The operator can configure the timer controlling the inactivity of the UE in the RNC, the default inactivity threshold being normally in order of dozens of minutes, e.g. 30 minutes. The inactivity detection function of the RNS (e.g. the RNC) may be based also on some other criteria, such as traffic volume control, traffic measurement, RLC buffers, timers etc. A UE that is in the idle state will need more time to set up a new data connection. This is because the set up procedure involves more signaling (e.g. RRC). The time needed to go from idle state to active state (CELL_PCH) is more than five seconds, and to CELL_DCH typically more than 10 seconds.

Five second setup time to go from the Idle state to the CELL_PCH is not an issue for end-users using data services such as FTP, web browsing, MMS, etc. This is mainly because these services can tolerate some extra delays if those are rare enough. However, for a PoC user that is logged on to a PoC session, five extra seconds start-to-talk time or delay as compared with the other RRC states is definitely too much. The start-to-talk delay may be defined as the time since the PTT button is pressed until the start-to-speak indication is given to the user (the user can start speaking). According to PoC service usability studies performed, the start-to-speak delay of 4 to 5 seconds is still experienced as annoying. Delay of 1 s or less would not be noticed at all. Delay less than 3 seconds can be considered of a reasonable quality. Thus, there is a need to reduce the communication setup delays.

Referring now to Figures 5 and 6, an example of a first aspect of the present invention will be described. An inactivity timer T1 is provided in a me-

dia communication (e.g. PoC) server for an ongoing real time media (e.g. PoC) session. Each time an activity (e.g. PoC data) is detected in the session (step 61 in Figure 6), the inactivity timer T1 is reset (step 62). If no activity has been noticed in a session for some predefined time T1 (step 63), then the server 14 will send dummy traffic (e.g. data or message) to all UE 2,3 that belong to this session (step 64). The dummy traffic, when received in the radio access network RNS 5,6, will reset (steps 51,52 in Figure 5) the inactivity or idle timer(s) T2,T3 controlling the transition from the CELL_PCH state (in more general terms, from a common channel state) to an Idle state. As a result, the UEs 2,3 that are logged on to real time (e.g. PoC) sessions are prevented from going into an idle state and can be kept always in active states. There are several advantages in this solution. Firstly, no (parameter) change is needed in the (e.g. WCDMA) radio networks. For example, the idle timer T2,T3 can be configured as default. Amount of dummy data sent is small because the UE typically go to the idle state after a relatively long period T2,T3 (e.g. 30 minutes) of inactivity. Therefore, the real time media (e.g. PoC) server can send dummy packets at relatively long intervals T1<T2,T3, for instance every 25 minutes, if no activity has been detected in one session. A UE that disconnects from a real-time (e.g. PoC) session will not receive dummy packets and therefore may go to idle state as normally if it is inactive long enough. Dummy packets may also be sent from server to UEs that are not using access network (e.g. WCDMA radio networks) that do not contain the idle timer (e.g. GPRS-GSM or WLAN or LAN). In such cases this method do not improve the performances at all. However, it does not decrease the end-to-end service performance either since these dummy packets will not affect on end-to-end.

In an embodiment of the invention, the issue above is overcome such that the UEs that are in access networks (e.g. WCDMA) notify the server (e.g. by sending their dummy packets or any other packet) that they need to receive dummy data in order to keep them in an active state. As a consequence, the server knows to which UE it should send dummy packets. Any system performance degradation in e.g. GPRS networks is avoided. For example, in Figure 1, if all UEs 1,2,3,4 are logged in a PoC session, all except UE1 that is located in the BSS/GSM access network would receive dummy data from the PoC server.

According to an embodiment of the present invention, a support node in a packet switched core network 10,11 that provides a real-time media

connection to a user equipment UE, is configured to send, during inactive periods of the real-time media connection, dummy media towards the user equipment, in order to reset an inactivity timer of a common channel state in the radio access network and to thereby prevent the respective user equipment from going to an idle state. In other words the functionality described above regarding the PoC server (Figures 5 and 6) is implemented in the support node. When the packet switched core network is a GPRS (General Packet Radio Service) type core network, the support node comprises a serving GPRS service node or a gateway GPRS service node. The SGSN or GGSN can determine that a flow accessing a certain Access Point will benefit from receiving from time to time some dummy data in order to wake the UE(s) up. The SGSN also knows which radio access technology a UE is using and can therefore send dummy data e.g. to WCDMA terminals only but will not send dummy data to UEs located in e.g. GSM BSS.

Referring now to Figures 7 and 8, examples of a second aspect of the present invention will be described.

As noted above, communication setup delays are very critical performance indicators of the PoC service and other corresponding real time media communication. A DCH (dedicated radio channel) will need to be established for PoC service at least for carrying voice data traffic. DCH establishment delays are around 1 second from the active states. The DCH establishment delays can be quite annoying during a PoC conversation especially because DCH delays are counted for each UE so total end-to-end delay is up to 2*DCH setup time.

It should be appreciated that a UE that is e.g. in cell_PCH state will not go to cell_DCH (i.e. establish a DCH) whenever it has data to send. The UE measures the amount of data to be transmitted in the transmission buffer in the UE, and reports the buffer status to the radio network controller RNC in the RNS in order to assist a dynamic radio bearer control. The measurement parameters can be set by the RNC. Measurement reports can be triggered using two different mechanisms, periodical and event triggered. The reporting criteria are specified in the measurement control message and may include one or more of Buffer Occupancy, Average of Buffer Occupancy, and Variance of Buffer Occupancy. The UE performs measurements and transmit measurement reports according to the measurement control information. For the uplink data transmission, the UE reports the observed traffic volume to the network in

order for the network to re-evaluate the current allocation of resources. This report contains e.g. the amount of data to be transmitted or the buffer status in the UE. The traffic volume or the buffer status depends on the activity of higher layer functions in the UE. For example, in the PoC service, the operation of a speech codec in the UE may be such that when a voice activity detector (VAD) indicates silence (and/or the user does not press the tangent), the speech codec does not provide any data to the access network (e.g. to the RLC buffer) in the UE, not even silence indicator frames, which are generated during a conventional voice supporting the discontinuous transmission (DTX).

When the user of the UE wants to say something to the other member(s) in the PoC call, she/he presses the tangent in the UE. The tangent button activates the speech codec regardless of the voice activity and speech codec starts to generate data into the RLC buffer in the UE. When the UE is in a common channel state (e.g. CELL_FACH), it reports the event to the RNC, which activates the transition to CELL_DCH state.

The RNC allocates the required capacity (including a DCH), detects a need to change the RLC parameters, carries out a radio link setup procedure with a base station BS, and commands the UE to the CELL_DCH state.

Similarly, the RNC may detect a capacity need in the downlink direction (e.g. based on traffic volume measurements, the downlink buffer status, etc.) and activate the transition to CELL_DCH state itself.

The amount of data sent by the UE needs to be large enough (128 bytes by default) and therefore signaling messages sent during start to talk procedure may not be big enough to trigger DCH. These messages may instead be sent over RACH and FACH. The amount of data needed to trigger DCH is configurable but again optimal values may not be selected according to one service only in the radio access network RNS.

Thus, according to another aspect of the present invention, a media communication server (e.g. the PoC server), and possibly a sending UE, is forced to send enough data so that set up of the DCH is triggered during the start to talk procedure. This dummy data is preferably sent immediately after sending the actual signaling message relating to the initiated start-to-talk procedure. As noted above, signalling in the PoC environment comprises Session Initiation Protocol (SIP) messages and Real-time Transport Control Protocol (RTCP) messages. Messages typically related to the start-to-speak situation

include SIP REFER request, SIP INVITE request, RTCP Floor Request, and RTCP Floor taken message.

Compressed SIP signalling messages are a bit more than 200 bytes whereas RTCP floor granted/taken/request messages are less than 100 bytes. In an embodiment of the invention, the amount of dummy data sent immediately after sending the signalling message (e.g. SIP REFER or RTCP FLOOR REQUEST or RTCP FLOOR TAKEN) is such that the total amount of data (signaling message + dummy data) is large enough to trigger DCH for the sending UE and/or the receiving UE(s). As a consequence, the amount of dummy data can be quite small: tens of bytes in case of SIP message sent and around 150 bytes in case of RTCP message. The aim of the invention is thus achieved with the minimum overhead data sent, and the capacity of the system is not wasted due to the invention.

Figure 7 shows a signaling diagram illustrating an example of starting a real-time media transaction, e.g. a PoC speech, in accordance with an embodiment of the present invention. Figure 8 illustrates an example of the operation of the UE in accordance with the principles of the present invention. Figure 9 illustrates an example of the operation of a PoC server in accordance with principles of the present invention.

Let us assume that the user equipment UE2 is initially in CELL_FACH state or CELL_PCH state. When the user of the UE2 wants to say something to the other member(s) in the PoC session, she/he presses the pressel PTT 306 in the UE. The PoC application 301 in the controller 305 detects that the pressel PTT is pressed (step 81 in Figure 8), and generates a SIP REFER message (or SIP INVITE) and transfers it to the RLC transmit buffer in the UE2 (steps 82 and 83). Further, in accordance with an embodiment of the invention, the PoC application (or e.g. the speech codec) also generates dummy data which is also transferred to the RLC buffer in the UE2 (step 84). The amount of the dummy data is such that the total data level in the RLC buffer will exceed the DCH threshold (e.g. 200 bytes). As a result, the UE2 will initiate a RRC procedure to setup a Dedicated Channel (DCH), e.g. the UE2 sends a capacity request (e.g. an RRC MEASUREMENT REPORT message) to the RNC, which activates a DCH for the PoC session. Now the UE2 is able to send the content of the RLC buffer, i.e. the signalling message and the dummy data, over the DCH to the RNS 5, and further to the PoC Server. Upon receiving the SIP REFER message, and if the turn to speak (a

speech item) is granted to the UE2, returns a RTCP Floor granted message. Upon receiving the RTCP Floor granted message, the UE2 will give a Start-to-speak indication (such as a beep) to the user, and the user starts to speak. The Voice activity detector VAD will detect the speech and start to generate speech data stream to the RLC buffer. This data can be sent immediately since the DCH is already set up. However, the start-to-speak time (time from pressing the pressel to the start-to-speak indication) is prolonged in comparison with the present case, since the DCH is setup therebetween.

In an alternative embodiment of the invention, the UE first waits (after the step 83 in Figure 8) that the signalling message (SIP or RTCP) is fully transmitted over the radio before sending the dummy data that would trigger DCH set up. Now the amount of the dummy data must alone exceed the triggering threshold. In this approach, the sending of the actual start-to-speak message (such as the SIP TRANSFER) and the response (such as the RTCP Floor granted), and thereby the start-to-speak indication, are not delayed due to the DCH setup. On the other hand, the DCH setup will still be initiated before the user starts to speak. Thus, this approach allows the start-to-speak time to remain below 1 second whereas the conversation delay would be decreased in comparison with the case where the dummy data is not sent.

In still another embodiment, the sending UE2 does not send any dummy data. In that such case, the reduction of the delay will be achieved by the operation of the PoC server towards the receiving side, as will be explained below.

Upon receiving the initial start-to-speak message (such as the SIP TRANSFER) from and having sent the response (such as the RTCP Floor granted) to the UE2 (steps 91 and 92 in Figure 8), the PoC server will send an appropriate signaling message (such as the RTCP Floor taken) to the UE3 (step 93). In an embodiment of the invention, the PoC server also sends dummy data with or immediately after the actual signaling message (step 94). The amount of data is such that it alone or together with the signaling message exceeds the DCH setup threshold in the serving RNS6 of the receiving UE3. As a consequence, a downlink DCH is setup for the UE3 and the signaling message and the dummy data are sent to the UE3 over the established DCH. Since the DCH is now ready, the subsequent RTP voice stream from the UE2 can be sent to the UE3 without the DCH setup delay. Thus, the conversation

delay will be decreased significantly, typically over 1 second (the DCH setup delay of the receiving user).

The inventive operation of the PoC server for the receiving side may be applied alone or in combination with any of the above operations of the sending UE. It should be noted that although the sending UE would not send any dummy data to trigger the DCH setup, the server would still decrease speech round trip time delay by sending to the receiving UE a dummy message immediately after sending the RTCP FLOOR TAKEN message. This server only solution may in fact be advantageous because it would allow to keep the start-to-speak time under 1 second (DCH establishment delay not counted) whereas the Speech round trip time would be decreased significantly.

Figure 10 shows a signaling flow diagram illustrating an example of another session event to which the principles of the present can be applied. The present speaker, e.g. the user of UE2, stops speaking. This is indicated by release of the PTT pressel, for instance. The UE2 signals the "stop of talk" event to the PoC server. This signaling may include a RTCP Floor Release, for example. In response to receiving the "stop of talk" signaling from the PoC server indicates the event to the receiving UE3 by means of an appropriate signaling. This signaling may include a RTCP Floor Idle message. The UE3 indicates the event (e.g. Floor Idle) to the user by an appropriate indication, such as a beep. After a delay caused by the human reaction, the user presses the PTT pressel. This may cause a similar procedure as described with respect to Figure 7. The PoC application 301 in the controller 305 detects that the pressel PTT is pressed, and generates an appropriate message (such as a RTCP Floor Request) and transfers it to the RLC transmit buffer in the UE3. Further, in accordance with an embodiment of the invention, the PoC application (or e.g. the speech codec) also generates dummy data which is also transferred to the RLC buffer in the UE3. The amount of the dummy data is such that the total data level in the RLC buffer will exceed the DCH threshold (e.g. 200 bytes). As a result, the UE2 will initiate a RRC procedure to setup a Dedicated Channel (DCH), and the RNC in the RNS6 activates a DCH for the PoC session. Now the UE3 is able to send the content of the RLC buffer, i.e. the signalling message and the dummy data, over the DCH to the RNS 6, and further to the PoC Server. Upon receiving the message, and if the turn to speak (a speech item) is granted to the UE3, the PoC server returns an appropriate response, such as a RTCP Floor granted message. Upon receiving the

RTCP Floor granted message, the UE3 will give a Start-to-speak indication (such as a beeb) to the user, and the user starts to speak after a human reaction time. The Voice activity detector VAD will detect the speech and start to generate speech data stream to the RLC buffer. Again, the voice data can be sent immediately since the DCH is already set up.

Alternatively, similar to one of the embodiments described above, the UE3 may send the actual message, e.g. RTCP Floor Granted message, first and the dummy data later. Still further, the UE3 may send only the actual message, e.g. RTCP Floor Granted message.

Upon receiving the initial message (such as the RTCP Floor Request) from and having sent the response (such as the RTCP Floor granted) to the UE3, the PoC server will send an appropriate signaling message (such as the RTCP Floor taken) to the UE2. In an embodiment of the invention, the PoC server also sends dummy data with or immediately after the actual signaling message. The amount of data is such that it alone or together with the signaling message exceeds the DCH setup threshold in the serving RNS5 of the UE2. As a consequence, a downlink DCH is setup for the UE2 and the signaling message and the dummy data are sent to the UE2 over the established DCH. Since the DCH is now ready, the subsequent RTP voice stream from the UE3 can be sent to the UE2 without the DCH setup delay.

In all embodiments relating to the second aspect of the invention, the PoC server may selectively send dummy data only to the UEs which are located in appropriate access networks (such as WCDMA), as discussed above relating to the first aspect of the invention.

In all embodiments of the invention, the serving access networks of the sending UE and the receiving UE(s) may be same one or different ones.

It should be appreciated that all the above operation beyond sending the dummy data for triggering the DCH setup or resetting inactivity timer may basically be implemented in accordance with the 3GPP specifications and existing PoC functionality.

Various embodiments of the invention have been described, but it will be appreciated by persons skilled in the art that these embodiments are merely illustrative and that many other embodiments are possible. The intended scope of the invention is set forth by the following claims, rather than the preceding description, and all variations that fall within the scope and spirit of the claims are intended to be embraced therein.

L 3

CLAIMS

1. Method of controlling a real-time media session, comprising

sending first signalling from a first user equipment via a serving access network of the first user equipment to a first media communication server in response to user's action during an established real-time media session,

sending second signalling from the first media communication server towards the first user equipment,

sending third signalling from the first media communication server towards second user equipment,

sending, immediately following the first and/or the second and/or the third signalling, dummy media traffic from the first media communication server towards the first and second user equipments, in order to trigger a dedicated channel setup for the first and/or second user equipments in their respective serving access network prior to an actual user media stream from the first user equipment begins.

2. A method according to claim 1, comprising

setting the amount of the dummy data such that the dummy data and the first signalling data together exceeds a threshold level for triggering the dedicated channel setup.

3. A method according to claim 1 or 2 comprising,

sending, immediately following the first, second and/or third signalling, dummy media traffic only if the session inactivity time prior the first signalling exceeded a certain threshold.

4. A method according to claim 1, 2 or 3 for a packet-mode voice communication, comprising

sending said first signalling in response to detecting, in the first user equipment, activation of a push-to-talk pressel or a like.

5. A method according to any one of preceding claims, wherein said first and/or second signalling comprises a Session Initiation Protocol (SIP) message and/or a Real-time Transport Control Protocol (RTCP) message, preferably one or more of: SIP REFER request, SIP INVITE request, RTCP Floor Request, and RTCP Floor taken message.

6. A method according to any one of claims 1-5, wherein the real-time media service is a push-to-talk over cellular or corresponding packet-mode voice communication service of a client-server type, the real-time media

stream is packet-mode speech, and/or at least one of the serving access networks comprise a radio access network of a wideband code division multiple access type.

7. A method of controlling a real-time media session, comprising

establishing a real-time media session between first user equipment and second user equipment via a serving access network of the first user equipment, via at least a first media communication server, and via a serving access network of the second user equipment,

sending, by the media communication server or a support node in a packet switched core network during inactive periods of the real-time media session, dummy media towards at least one of the first and second user equipment, in order to reset an inactivity timer of a common channel state in the serving access network of the respective user equipment and to thereby prevent the respective user equipment from going to an idle state.

8. A method according to claim 7, comprising

monitoring media activity of the real-time media session in the first media communication server or in the support node,

if no media activity is detected in the real-time media session for a predetermined period of time, sending said dummy media traffic from the first media communication server or the support node towards at least one of the first and second user equipment.

9. A method according to claim 7 or 8, comprising sending said dummy media traffic to said at least one of the first and second user equipment only if the respective user equipment is located in an access network in which a dedicated channel setup can be triggered by a dummy media traffic.

10. A method according to claim 9, comprising notifying, by the respective user equipment, that it is located in an access network in which a dedicated channel setup can be triggered by a dummy media traffic.

11. A method according to any one of claims 7-10, wherein the real-time media service is a push-to-talk over cellular or corresponding packet-mode voice communication service of a client-server type, the real-time media stream is packet-mode speech, and/or at least one of the serving access networks comprise a radio access network of a wideband code division multiple access type.

12. A method according to any one of claims 7-11, wherein the packet switched core network is a GPRS (General Packet Radio Service) type

core network, and wherein the support node comprises a serving GPRS service node or a gateway GPRS service node.

13. A media communication server for providing real-time media sessions between user equipments located in one or more access networks, wherein

the media communication server is configured to receive first signalling sent by a first user equipment via a serving access network of the first user equipment in response to user's action during an real-time media session established between the first user equipment and a second user equipment,

the media communication server is configured to send second signalling towards the first user equipment upon receiving said first signalling,

the media communication server is configured to send third signalling towards the second user equipment upon receiving said first signalling,

the media communication server is configured to send, immediately following the first, second, and/or third signalling, dummy media traffic towards the first and/or second user equipment, in order to trigger a dedicated channel setup for the first and/or second user equipment in a respective serving access network prior to an actual user media stream from the first user equipment begins.

14. A media communication server according to claim 13, wherein said first and/or second signalling comprises a Session Initiation Protocol (SIP) message and/or a Real-time Transport Control Protocol (RTCP) message, preferably one or more of: SIP REFER request, SIP INVITE request, RTCP Floor Request, and RTCP Floor taken message.

15. A media communication server according to claim 13 or 14, wherein the media server is arranged to send said dummy media traffic from the first media server to the first and/or second user equipment only if these user equipments are located in an access network in which a dedicated channel setup can be triggered by a dummy media traffic.

16. A media communication server according to claim 13, 14 or 15, wherein the real-time media service is a push-to-talk over cellular or corresponding packet-mode voice communication service of a client-server type, the real-time media stream is packet-mode speech, and/or at least one of the serving access networks comprise a radio access network of a wideband code division multiple access type.

17. A media communication server according to any one of claims 13-16, wherein the media communication server is configured to send dummy media traffic to first and/or second user equipment only if the session inactivity prior to first signalling exceeded a certain threshold, in order to limit the amount of unnecessary dummy data sent.

18. A media communication server for providing real-time media sessions between user equipments located in one or more access networks, wherein

the media communication server is configured to establisha real-time media session between first user equipment and second user equipment via a serving access network of the first user equipment and via a serving access network of the second user equipment,

the media communication server is configured to send, during inactive periods of the real-time media session, dummy media towards at least one of the first and second user equipment, in order to reset an inactivity timer of a common channel state in the serving access network of the respective user equipment and to thereby prevent the respective user equipment from going to an idle state.

19. A media communication server according to claim 18, wherein the media communication server is configured to monitor media activity of the real-time media session in the first media communication server or in the support node, and if no media activity is detected in real-time media session for a predetermined period of time, to send said dummy media traffic.

20., A media communication server according to claim 18 or 19, wherein the media server is arranged to send said dummy media traffic from the first media server to the second user equipment only if the second user equipment is located in an access network in which a dedicated channel setup can be triggered by a dummy media traffic.

21. A media communication server according to claim 18, 19 or 20, wherein the real-time media service is a push-to-talk over cellular or corresponding packet-mode voice communication service of a client-server type, the real-time media stream is packet-mode speech, and/or at least one of the serving access networks comprise a radio access network of a wideband code division multiple access type.

22. A support node for a packet switched core network, wherein

the support node is configured to establish a real-time media con-

nection between a user equipment located in a radio access network, and a media communication server,

the support node is configured to send, during inactive periods of the real-time media connection, dummy media towards the user equipment, in order to reset an inactivity timer of a common channel state in the radio access network and to thereby prevent the respective user equipment from going to an idle state.

23. A support node according to claim 22, wherein the real-time media service is a push-to-talk over cellular or corresponding packet mode-voice communication service of a client-server type, the real-time media stream is packet-mode speech, and/or at least one of the serving access networks comprise a radio access network of a wideband code division multiple access type.

24. A support node according to claim 22 or 23, wherein the packet switched core network is a GPRS (General Packet Radio Service) type core network, and wherein the support node comprises a serving GPRS support node or a gateway GPRS support node.

25. User equipment for a communication system, wherein

the user equipment is configured to establish a real-time media session via an access network and a media communication server,

the user equipment is configured to send a first signalling via the access network to the media communication server in response to user's action during the established real-time media session, and

the user equipment is configured to send, immediately following the first signalling, dummy media traffic to the media communication server, in order to trigger a dedicated channel setup for the user equipment in the access network of the first user equipment prior to an actual user media stream begins.

26. User equipment according to claim 25 for a packet-mode voice communication, wherein the user equipment is configured to send said first signaling when detecting activation of a push-to-talk pressel or a like.

27. User equipment according to claim 25 or 26, wherein said first signalling comprises a Session Initiation Protocol (SIP) message and/or a Real-time Transport Control Protocol (RTCP) message, preferably one or more of: SIP REFER request, SIP INVITE request, and RTCP Floor Request.

28. User equipment according to claim 25, 26 or 27, wherein the real-time media service is a push-to-talk over cellular or corresponding packet-mode voice communication service of a client-server type, the real-time media stream is packet-mode speech, and/or the access network comprises a radio access network of a wideband code division multiple access type.

29. User equipment according to any one of claims 25-28, wherein the amount of the dummy data is such that the dummy data and the first signalling data together exceeds a threshold level for triggering the dedicated channel setup.

30. User equipment according to any one of claims 25-29, wherein the user equipment is configured to keep the first signalling and the dummy data in a transmission buffer until the triggered dedicated channel setup has been completed, and to send the first signalling and the dummy data over the dedicated channel.

31. User equipment according to any one of claims 25-29, wherein the user equipment is configured to send the first signalling completely before sending the dummy data and triggering the dedicated channel setup.

32. User equipment according to any one of claims 25-29, wherein the user equipment is configured to send dummy media traffic to the media communication server only if the session inactivity time prior sending the first signalling exceeds a certain threshold, in order to limit the amount of unnecessary dummy data sent.

L4

/

(57) Abstract

A real-time media session is between user equipment and a media communication server via a serving access network. According to the Invention, dummy data (e.g. a dummy message) is sent in order to maintain a dedicated channel during inactive periods of a real-time media session or to trigger an early setup of a dedicated channel in the access network. Thereby, user equipments that are logged on to a real-time media (e.g. PoC) session are prevented form going to a radio resource idle state, and therefore potential long extra delays during real time media (e.g. PoC) service usage avoided. The invention further allows the sending and receiving user equipments to set up their dedicated channels (DCH) already during the start to talk procedure of the transmitting user equipment, which in turn potentially reduces end-to-end delays during the conversation.

L5

1

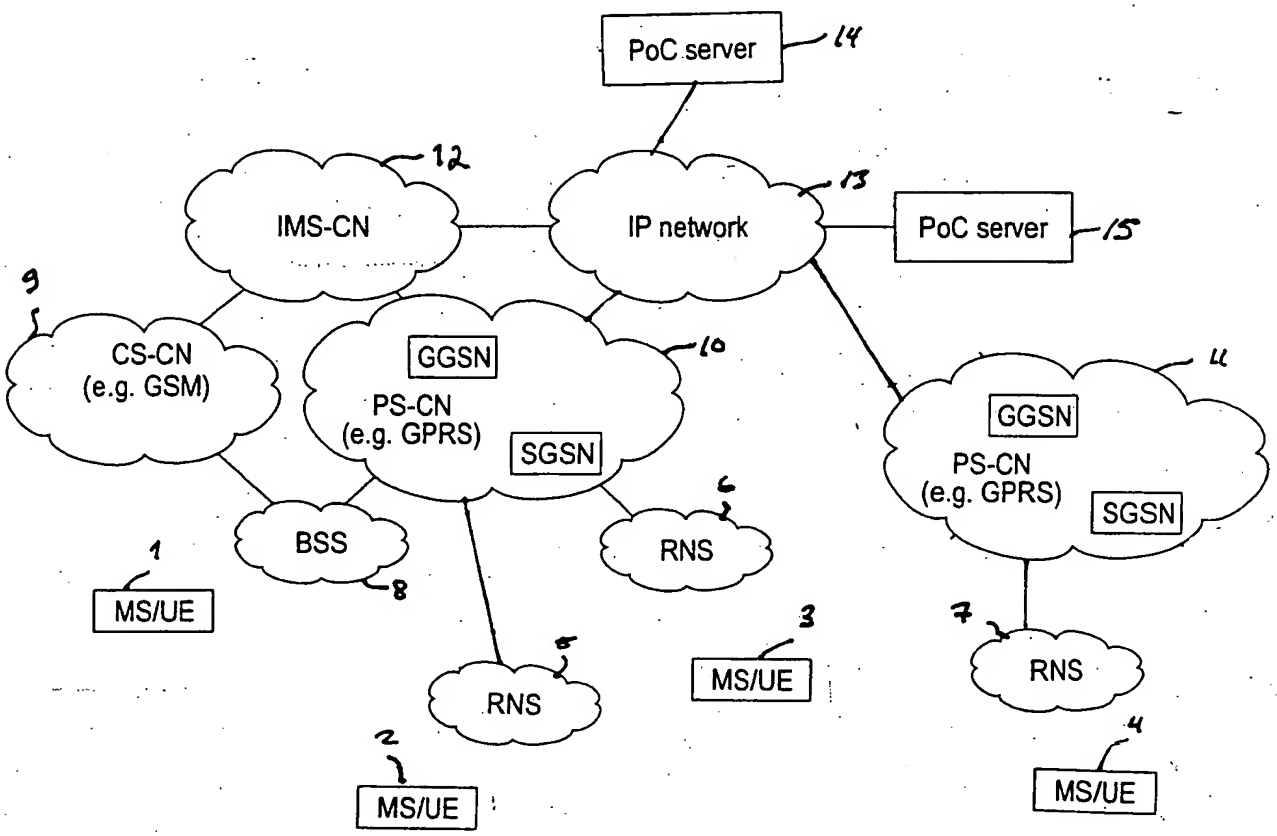


Fig. 1

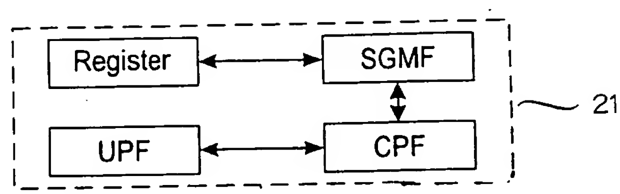


Fig. 2

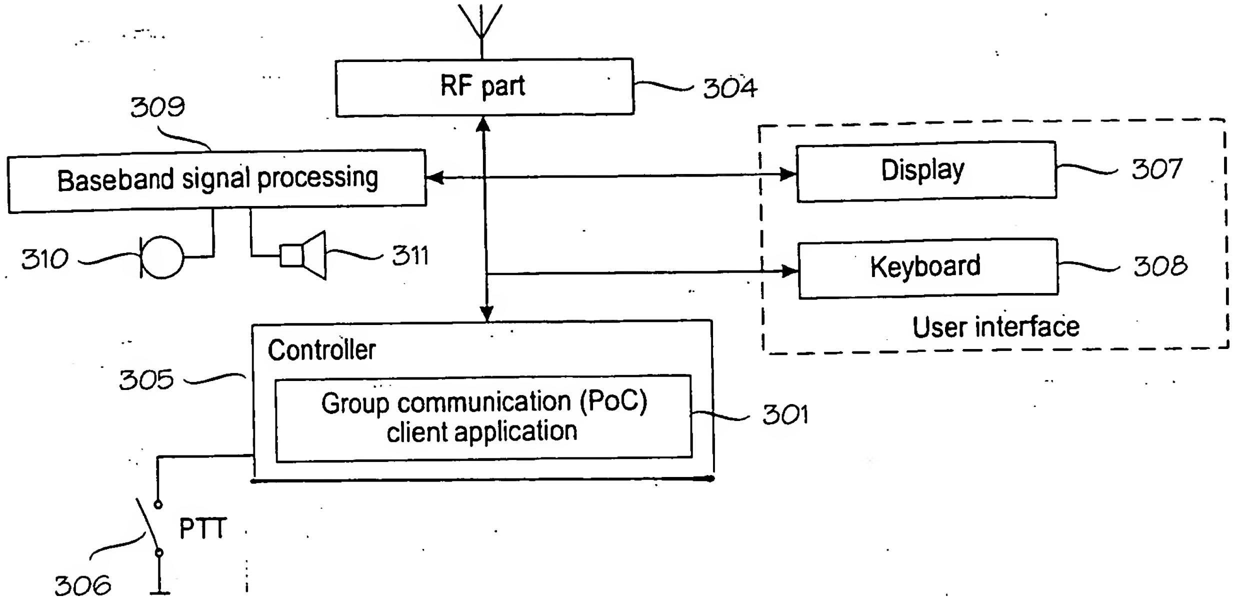

RF part
304
309
Baseband signal processing
Display
307
Keyboard
308
User interface
310
311
Controller
305
Group communication (PoC) client application
301
PTT
306

Fig. 3

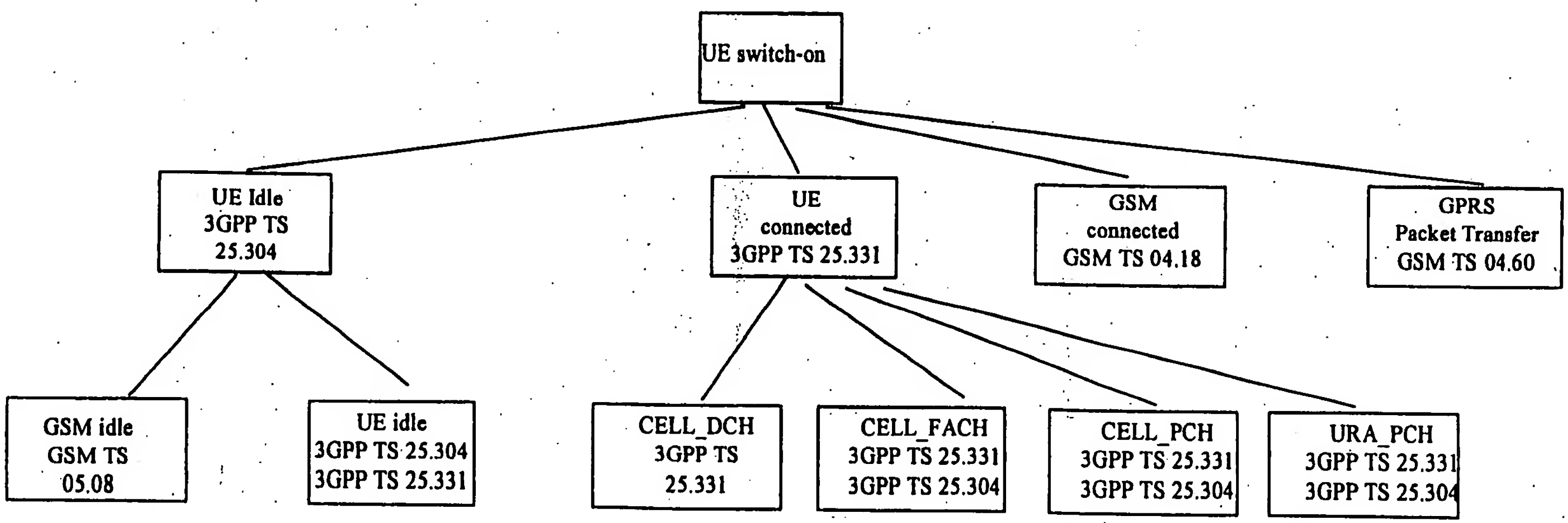

UE switch-on
UE Idle 3GPP TS 25.304
UE connected 3GPP TS 25.331
GSM connected GSM TS 04.18
GPRS Packet Transfer GSM TS 04.60
GSM idle GSM TS 05.08
UE idle 3GPP TS 25.304 3GPP TS 25.331
CELL_DCH 3GPP TS 25.331
CELL_FACH 3GPP TS 25.331 3GPP TS 25.304
CELL_PCH 3GPP TS 25.331 3GPP TS 25.304
URA_PCH 3GPP TS 25.331 3GPP TS 25.304

Fig. 4

L5

3

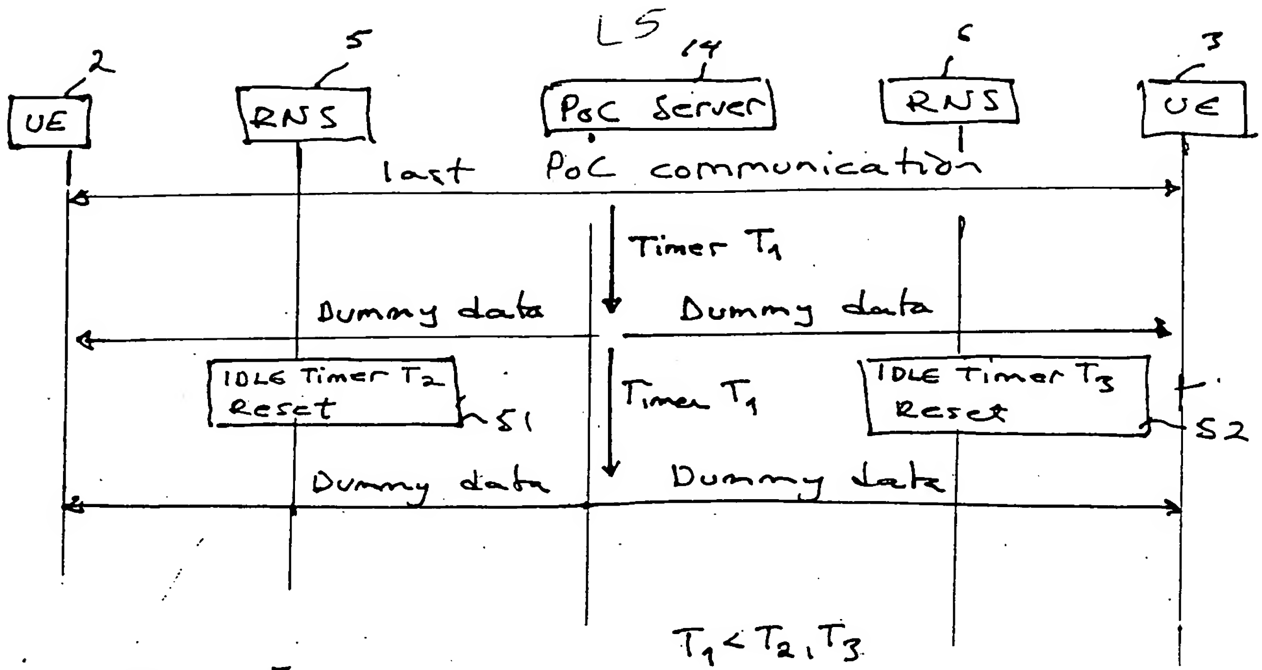

LS
14
PoC Server
2
UE
5
RNS
6
RNS
3
UE
last PoC communication
Timer T_1
Dummy data
Dummy data
IDLE Timer T_2 Reset
51
Timer T_1
IDLE Timer T_3 Reset
52
Dummy data
Dummy data
$T_1 < T_2, T_3$

Fig. 5

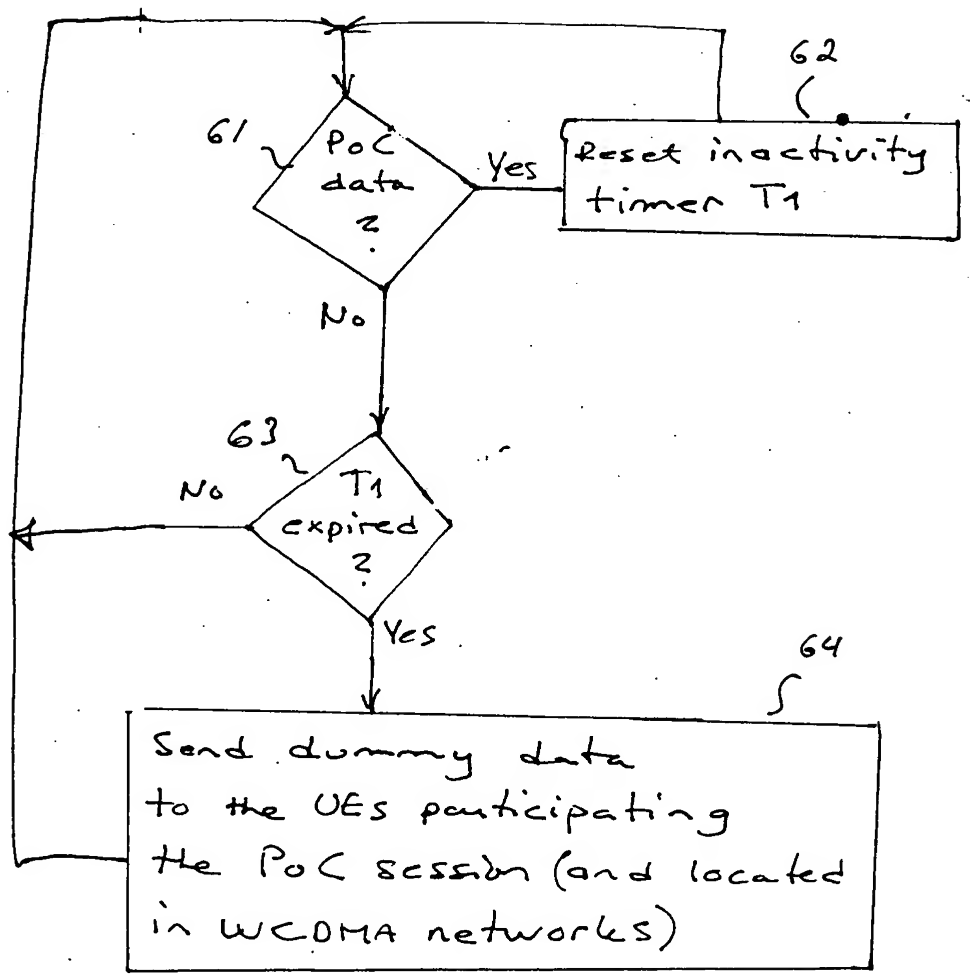

61
PoC data ?
Yes
62
Reset inactivity timer T1
No
63
T1 expired ?
No
Yes
64
Send dummy data to the UEs participating the PoC session (and located in WCDMA networks)

Fig. 6

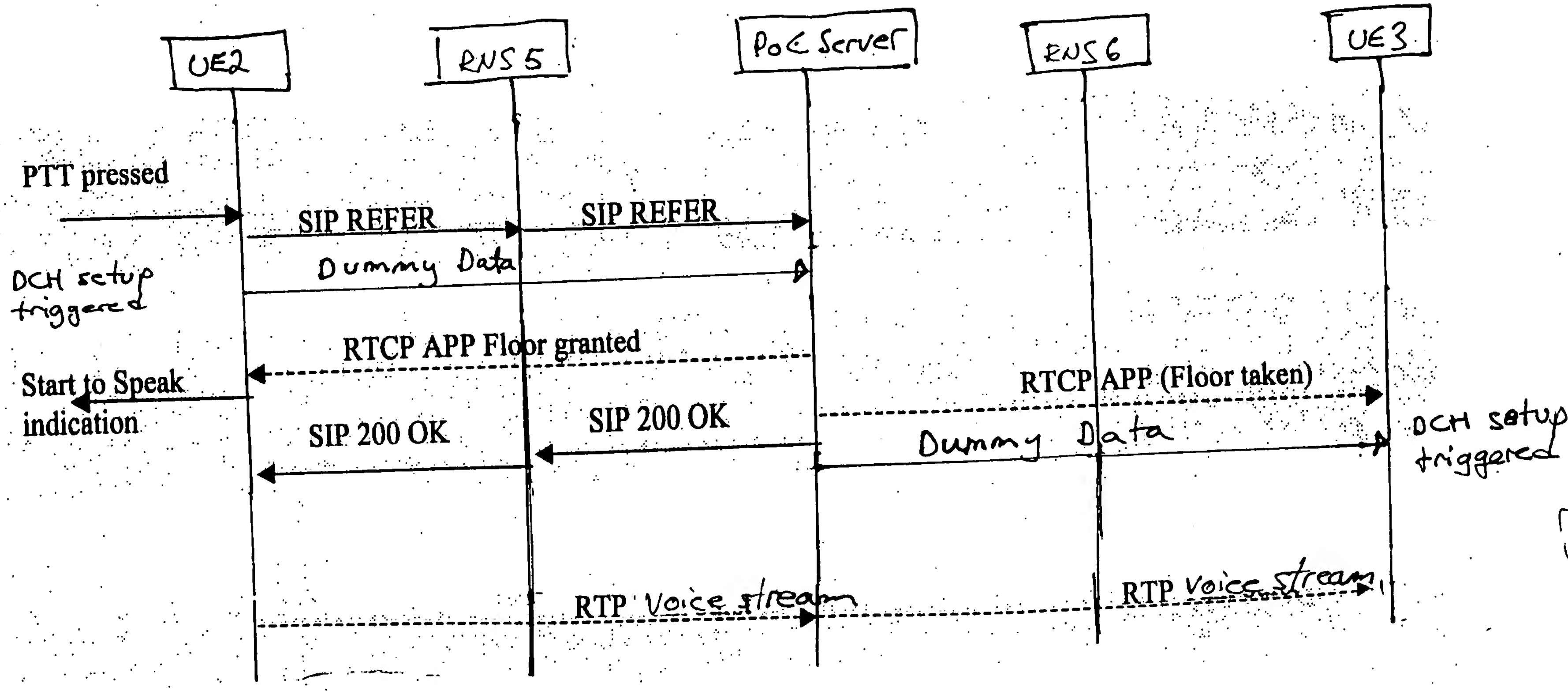

UE2
RNS 5
PoC Server
RNS 6
UE3
PTT pressed
SIP REFER
SIP REFER
Dummy Data
DCH setup triggered
RTCP APP Floor granted
Start to Speak indication
RTCP APP (Floor taken)
SIP 200 OK
SIP 200 OK
Dummy Data
DCH setup triggered
RTP Voice stream
RTP Voice stream

Fig. 7

L5

5

Fig. 8

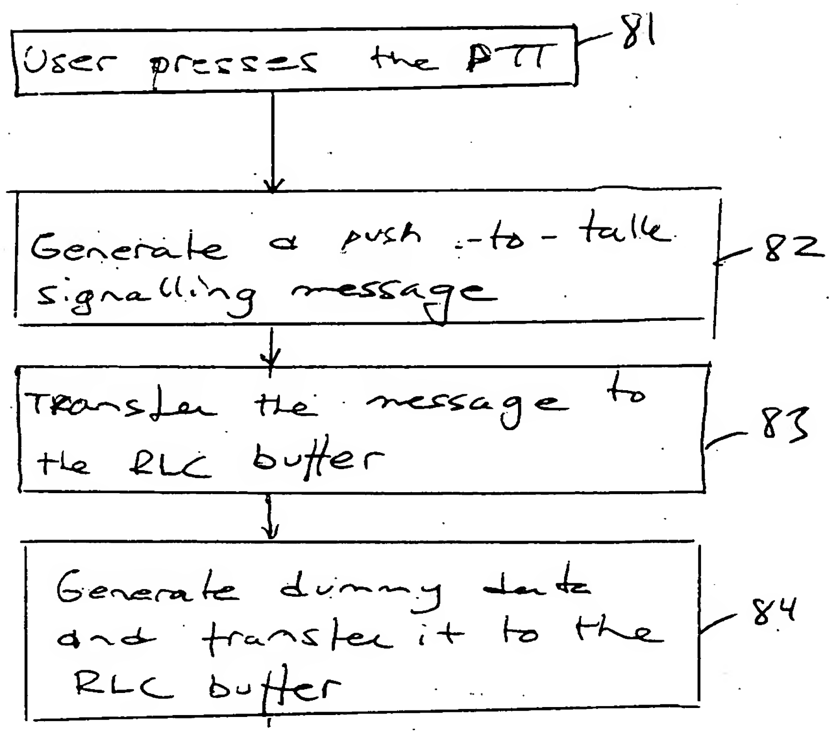
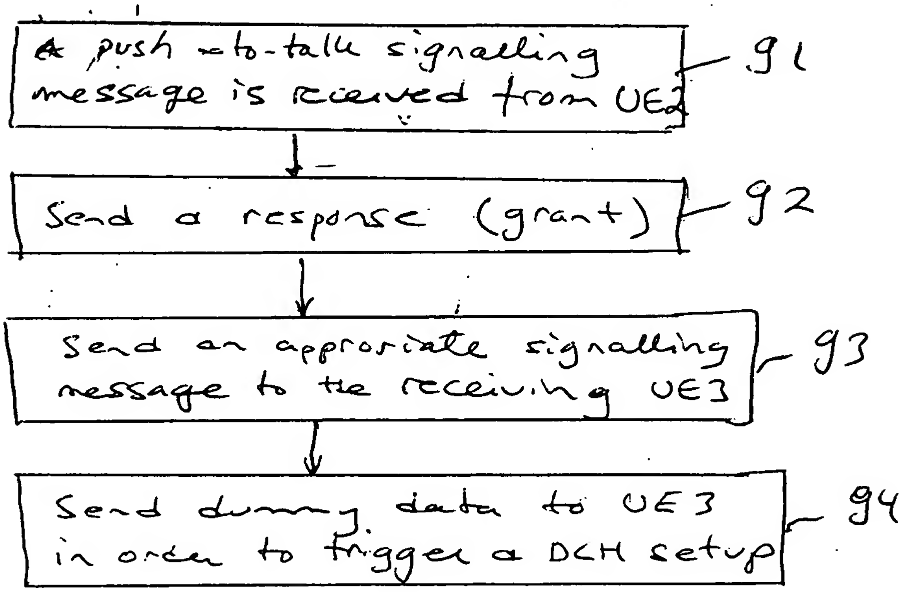

User presses the PTT
81
Generate a push-to-talk signalling message
82
Transfer the message to the RLC buffer
83
Generate dummy data and transfer it to the RLC buffer
84

Fig. 9.


A push-to-talk signalling message is received from UE2
91
Send a response (grant)
92
Send an appropriate signalling message to the receiving UE3
93
Send dummy data to UE3 in order to trigger a DCH setup
94

C5

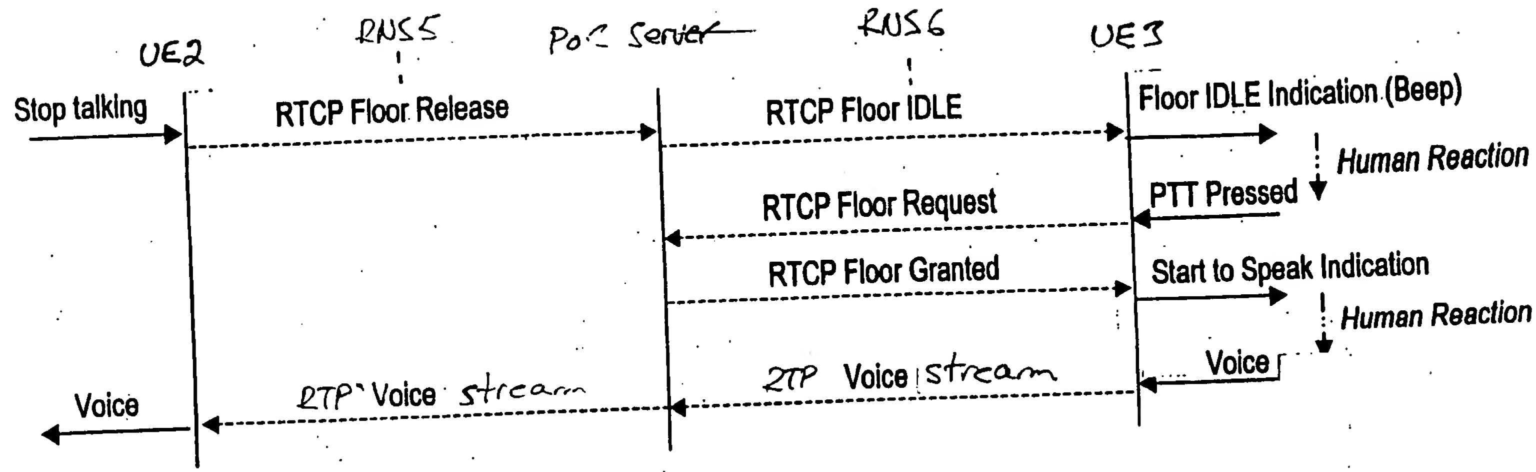

UE2
RNS5
PoC Server
RNS6
UE3
Stop talking
RTCP Floor Release
RTCP Floor IDLE
Floor IDLE Indication (Beep)
Human Reaction
PTT Pressed
RTCP Floor Request
RTCP Floor Granted
Start to Speak Indication
Human Reaction
Voice
RTP Voice stream
RTP Voice stream
Voice

Fig. 10.